CERTIFICATION OF TRANSLATION

I, ***Sohee Kim***, an employee of Y.P.LEE, MOCK & PARTNERS of Koryo Bldg., 1575-1 Seocho-dong, Seocho-gu, Seoul, Republic of Korea, hereby declare under penalty of perjury that I understand the Korean language and the English language; that I am fully capable of translating from Korean to English and vice versa; and that, to the best of my knowledge and belief, the statement in the English language in the attached translation of ***Korean Patent Application No. 10-2000-0009624*** consisting of *17* pages, have the same meanings as the statements in the Korean language in the original document, a copy of which I have examined.

Signed this 18th day of June 2008

Kimsohee

ABSTRACT

[Abstract of the Disclosure]

An apparatus and method of transceiving a bitstream, by which a video bitstream is stably transmitted by the effective use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream in a communication network, is provided. In this method, a source is encoded into a bitstream using a predetermined type of coding. Then, the encoded bitstream is transmitted to each communication protocol layer, while the header of each communication protocol layer is added to a payload. The header is transmitted separately from the bitstream.

[Representative Drawing]

FIG. 5

SPECIFICATION

[Title of the Invention]

Apparatus for transmitting/receiving bitstream in network and method thereof

[Brief Description of the Drawings]

FIG. 1 is a view illustrating a method of transmitting a video bit stream under an H.323 environment.

FIG. 2 is a view illustrating a method of transceiving a file between a server and a client in a communication network.

FIG. 3 is a bock diagram of a video stream transmitting apparatus according to the present invention.

FIG. 4 is a block diagram of an apparatus for relaying and receiving a video stream, according to the present invention/

FIG. 5 is a view illustrating a method of transmitting a video bitstream in a situation where a wireless network communicates with an Internet network.

[Detailed Description of the Invention]

[Object of the Invention]

[Technical field of the Invention and Related Art prior to the Invention]

The present invention relates to an apparatus for transmitting/receiving a bitstream in a network and a method thereof, and more particularly, to an apparatus for transmitting/receiving a bitstream, by which a video bitstream is stably transmitted in a network including a wireless network and an Internet network by the effective use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream and a method thereof.

In packet networks such as Internet, typically, two logical ports exist when a channel is set to achieve the communication between two spots. That is, a real time protocol (RTP) packet, which has been packetized by a request for comments (RFC) protocol via the Internet network, is transmitted via the hierarchical structure of a real time protocol/user datagram protocol/Internet protocol (RTP/UDP/IP) or a real time protocol/transmission control protocol/Internet protocol (RTP/TCP/IP).

The RTP/TCP/IP operates in the acknowledge mode and can transmit data stably, so that it usually transmits control information. The RTP/UDP/IP operates in the unacknowledged mode, and transmits video data which usually must be processed in real time.

Referring to FIG. 1, in a transmitting terminal, a video bit stream sequentially undergoes an application layer (video source codec), an RTP layer, an UDP/IP and TCP/IP layer, a radio link layer protocol (RLP) layer, a layer 2 (L2) and a layer 1 (L1). In each layer, header information is added to the video bit stream, and the video bit stream having header information is transmitted to a network. Here, the TCP transmits control information, and the UDP transmits an RTP packet. In a receiving terminal, the video bit stream undergoes an UDP/IP and TCP/IP layer, an RTP layer and an application layer (video source codec), and is decoded into video data.

As shown in FIG. 2, a server 200 divides a video bit stream into a high significant bitstream 210 and a low significant bitstream 220. The server 200 first transmits the high significant bitstream 210 to a client 230 and then transmits the low significant bitstream 220 to the client 230. Thus, a packet loss can be minimized.

First, a server 200 reads a video bit stream produced by a video source codec to divide the read video bit stream into a high significant bitstream 210 and a low significant bitstream 220 before an RTP packet is produced. Next, the server 200 transmits the high significant bitstream 210 to a client 230 via an RTP/TCP/IP, as indicated by arrow $\in$, and receives an acknowledgment representing that transmission of the high significant bitstream 210 has been completed without error, as indicated by arrow $\notin$. Then, the server 200 transmits the low significant bitstream 220 to the client 230 via an RTP/UDP/IP regardless of error as indicated by arrow $\angle$. The client 230 reconstructs the high significant bitstream 210 and the low significant bitstream 220 back into the original video bitstream syntax.

As described above, under a conventional Internet environment, a bitstream produced by video source coding such as MPEG-4 or H.263 is transmitted to the UDP/IP layer. The UDP/IP does not notify a transmitting terminal of an acknowledgment of the reception of a packet from a receiving terminal. In other words, a packet loss occurs under the Internet environment. Thus, the bitstream packet cannot ascertain that the receiving terminal receives all of data transmitted from the transmitting terminal. If a bitstream packet is transmitted under a communication environment in

which an Internet network communicates with a wireless network, the packet data may have bit error under the wireless environment. That is, the bitstream packet may have a loss on the Internet while undergoing the Internet environment and the wireless environment, and bit error may be generated even though the bitstream packet has been transmitted without loss. Here, the bitstream packet includes a packet header and a payload header. If a bit error is included in these headers, the receiving terminal cannot perform suitable decoding.

When a video bitstream is transmitted in real time, the transmitting terminal classifies the video bitstream on the basis of the significance, and transmits significant portions first and then less significant portions. At this time, the receiving terminal must delay data until significant portions are received first, so that real time processing of data is difficult. That is, if significant portions are consecutively transmitted in an Internet network and a wireless network, the network must be continuously stable. Also, if a bitstream is classified according to significance during video source coding, it cannot be reconstructed into a bitstream packet which confirms to a current RFC protocol. Furthermore, the server 200 and the client 230 must always perform a pre-process for classifying a bitstream according to the significance to transmit and receive the bitstream, and a post-process for reconstructing the received upper and lower significant bitstreams into the original bitstream. Also, the process for classifying a bitstream on the basis of the significance before packetizing can only be performed at a video codec level which has already known the syntax of the video bitstream.

Therefore, data communication in an Internet network combined with a wireless environment causes a packet loss and a bit error as described above, thus deteriorating the quality of an image.

[Technical goal of the Invention]

An objective of the present invention is to provide a method of transmitting a bitstream, by which a video bitstream is stably transmitted by the effective use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream in a packet network.

Another objective of the present invention is to provide an apparatus for transceiving a bitstream, by which a video bitstream is stably transceived by the effective

use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream in a packet network.

[Structure of the Invention]

To achieve the first objective, there is provided a method of transmitting a bitstream in a communication network, according to an embodiment of the present invention, the method including: (a) coding a source into a bitstream using a predetermined type of coding; (b) adding the header of each communication protocol layer to a payload while transmitting the bitstream coded in the step (a) to each communication protocol layer; and (c) transmitting the header separately from the bitstream received from the step (b).

To achieve the first objective, there is provided a method of transmitting a bitstream in a communication network, according to another embodiment of the present invention, the method including: (a) coding a source into a bitstream using a predetermined type of coding; and (b) adding the header of each communication protocol layer to a payload while transmitting the bitstream coded in the step (a) to each communication protocol layer, and separately transmitting the payload and the header.

To achieve the second objective, there is provided an apparatus for transmitting a bitstream in a communication network, according to an embodiment of the present invention, the apparatus including: an encoder for encoding source data into a bitstream using a predetermined type of coding; a protocol processing unit for adding the header of each communication protocol layer to a payload while transmitting the bitstream encoded by the encoder to each communication protocol layer; and a packet processing unit for transmitting the bitstream processed by the protocol processing unit in an unacknowledged mode protocol and transmitting the header information in an acknowledged mode protocol.

To achieve the second objective, there is provided an apparatus for relaying and receiving a bitstream in a communication network, according to another embodiment of the present invention, the apparatus including: an extractor for separately extracting payloads and header information which corresponds to the header of each layer while transmitting a bitstream received in a separate transmission protocol in the communication network to each layer; an error determination processing unit for determining whether the header information extracted by the extractor has error, and, if it i

s determined that the header information has error, requesting re-transmission; a bit stream re-organizing unit for re-organizing a bitstream using the header information extracted by the extractor, if it is determined that the header information has no error ; and a decoder for decoding a bitstream re-organized by the bitstream re-organizing unit.

Hereinafter, a preferred embodiment of the present invention will be describe d with reference to the attached drawings.

FIG. 3 is a bock diagram of a video stream transmitting apparatus according t o the present invention.

MPEG-4 or H.263, which are widely used as a video coding method, include v arious types of standardized headers. In case that data is transmitted in a real time protocol (RTP) while undergoing each layer of an Internet protocol or a wireless pro tocol, the transmission format is used.

Referring to FIG. 3, a video codec unit 310 encodes data to a bitstream using an application program such as H.323. A protocol processing unit 320 transfers the bit stream encoded by the video codec unit 310 to each layer of a communication p rotocol, and simultaneously adds the header of each protocol to a payload. A pack et processing unit 330 packetizes a bitstream processed by the protocol processing unit 320 and transmits the bitstream packet in a user datagram protocol (UDP), whic h is an unacknowledged mode transmission protocol, and transmits only header info rmation in a transmission control protocol (TCP), which is an acknowledged mode tr ansmission protocol. In another embodiment, the packet processing unit 330 trans mits a payload, among a bitstream processed by the protocol processing unit 320, in an unacknowledged mode transmission protocol, and transmits only the added hea der information in an acknowledged mode transmission protocol.

MPEG-4 or H.263, which are widely used as a video coding method, include v arious types of standardized headers. In case that data is transmitted in a real time protocol (RTP) while undergoing each layer of an Internet protocol or a wireless pro tocol, the transmission format is used. Accordingly, when this video coding method is used, the header of each layer is added to a payload header in each layer. Thu s, only when the header of each layer and the payload header are safe from a bit err or, a video codec 480 of a receiving terminal can perform suitable decoding.

FIG. 4 is a block diagram of an apparatus for relaying and receiving a video st ream, according to the present invention. Referring to FIG. 4, a packet extractor 41 0 transfers to each layer a bitstream packet received in an unacknowledged or ackn owledged mode transmission protocol, while separately extracting a payload and the header of each layer from the bitstream packet. An error determination processin g unit 412 determines existence or non-existence of an error in the header informati on extracted by the packet extractor 410. If it is determined that an error exists in th e header information, the error determination processing unit 412 requests re-transm ission. On the other hand, if it is determined that there are no errors in the header, a bit stream re-organizing unit 420 re-organizes a video bitstream using the header o f each layer extracted by the packet extractor 410. A video codec unit 430 decodes the bitstream re-organized by the bitstream re-organizing unit 420.

FIG. 5 is a view illustrating a method of transmitting a video bitstream in a situ ation where a wireless network is interlocked with an Internet network. Referring to FIG. 5, reference numeral 510 indicates a wireless terminal on a transmitting side in cluding several layers, reference numeral 560 indicates a base station including sev eral layers, and reference numeral 580 indicates an Internet terminal on a receiving side including several layers.

First, the wireless terminal 510 includes a video source codec layer 512, whic h corresponds to an application layer, at the top, and sequentially includes an RTP l ayer 514, an TCP/IP layer 516, a radio link layer protocol (RLP) layer 522, a layer 2 (L2), and a layer 1 (L1) 526. Here, a multimedia codec of an audio source codec ot her than the video source codec can be used as the application layer.

The video source codec layer 512 encodes a video source into a video bitstre am using a video source coding method such as MPEG-4 or H.263 to form a payloa d header 532 and a video payload 534 as shown in (a). Then, the RTP layer 514 fo rms a packet by adding a video payload 545 filled with video data, a payload header 544, and an RTP header 543, and the UDP/IP or TCP/IP layer 516 adds an IP head er 541 and an UDP or TCP header 542 to the formed packet, as shown in (b). The RTP layer 522 and the L2 layer 524 add an L2 header 551 and an RLP header 552 t o the packet (b) as shown in (c).

Next, a video bitstream to which the header of each layer is added is transmitt ed to the base station 560 including identical layers, through an UDP or TCP. The

video bitstream (c) including headers can be divided into a portion which is transmitted through the UDP, which is an UNACK transmission protocol, and a portion which is transmitted through the TCP, which is an ACK transmission protocol.

In a first embodiment of the first method, a video bitstream, to which header information is added, is transmitted in the UDP, and the header information is separately transmitted in the TCP. When a bitstream is transmitted only in the UDP, if header information included in the bitstream is damaged, it is difficult for a receiving side to process the bitstream. Hence, in order to prevent the packet loss, the wireless terminal 510 individually packetizes the header of each layer, that is, the L2 header 551, the RLP header 552, the UDP header 542, the payload header 544, the RTP header 543, the multimedia header, which are added to a video bitstream after the video bitstream passes through each layer. At the same time or when re-transmission is requested, the wireless terminal 510 stably transmits the packetized headers in a TCP. Here, the transmission in the TCP at the request of re-transmission is performed in units of IP packets or RLPs.

In a second embodiment, in order to solve delay that may occur in real time environment, a video bitstream to which header information is added is transmitted in a UDP, and the header information is separately packetized and transmitted in a UDP simultaneously or when re-transmission is requested.

In the second method according to an embodiment, a video bitstream is separated into a payload portion and a header portion, and these portions are separately packetized. The payload portion is transmitted in a UDP, and simultaneously, the header portion is separately transmitted in a TCP. In another embodiment, the payload portion is transmitted in a UDP, and simultaneously, the header portion is separately transmitted in a UDP. In still another embodiment, in order to reduce a transmission time, a bitstream packet except for a portion from which a bit error is removed by the TCP layer can be transmitted to a UDP layer.

In the third method, when a bitstream transmitted via the TCP layer is re-transmitted a small number of times, the channel of transmission is determined to be stable to some extent. Accordingly, a small bitstream transmitted via the UDP can be transmitted via the TCP.

The base station 560 relays the layers of a wireless protocol, that is, an RLP layer, an L2 and an L1, to the layers of an Internet protocol, an UDP layer, an IP laye

r and an L1 (or ATM), in order to tunnel a bitstream received from the wireless terminal 510. At this time, when re-transmission is requested, data transmitted in a TCP is re-transmitted in units of IP packets or RLPs.

The IWF 570 relays a bitstream, which has passed through the layers of the base station 560, that is, a UDP, an IP, and an L1, to an UDP or TCP, an IP and an L1, in order to interface with the Internet terminal 580.

The Internet terminal 580, which is a final receiving side, decodes a bitstream received from the IWF 570 through an L1 576, an IP layer 572, a UDP or TCP layer 566, an RTP layer 564, and a video source codec layer 562. The Internet terminal 580 can properly decode a video bitstream which probably has a bit error, using an error resilient tool of video coding, referring to a payload and separately-received header information, when a packet received via the UDP layer has a bit error.

The present invention is not limited to the aforementioned embodiment, and it is apparent that modifications to this particular embodiment may be effected by those skilled in the art without departing from the spirit of the present invention. The present invention can also be applied to a case where an audio source codec other than a video source codec, or a source codec having the same function as the audio source codec has an error resilient tool with respect to a payload or a function which conforms to the error resilient tool.

Also, the above-described embodiment of the present invention can be written in a program that can be executed in computers, and can be realized in general-use digital computers which operate the program from a medium which is used in computers. The medium includes a magnetic storage medium (for example, a ROM, a floppy disc, a hard disc, and the like), an optical read-out medium (for example, a CD-ROM, a DVD and the like), and a storage medium such as a carrier wave (for example, transmission via the Internet).

[Effect of the Invention]

As described above, the present invention can be used in a case where a bitstream is bidirectionally communicated in real time or streamed in a one-way system in a packet network such as an Internet. Header information or the like is stably transmitted separately from a payload when a wireless network and an Internet network are linked, so that it can correct and check a bit error of a packet which has been pa

ssed through each layer. Also, in contrast with an existing method of transmitting data according to significance, a packet to which the present invention is applied can be processed independently of a video syntax. Furthermore, under an application environment where communication is achieved using a UDP, a bitstream including a bit error can be appropriately decoded by separately-received header information using an error resilient tool, that is, in cases of an adaptive multi rate (AMR) for an MPEG-4 audio mobile, an AMR for an UMTS, a speech codec, and the like.

What is claimed is:

1.

A method of transmitting a bitstream in a communication network, the method comprising:

(a) coding a source into a bitstream using a predetermined type of coding;

(b) adding the header of each communication protocol layer to a payload while transmitting the bitstream coded in the step (a) to each communication protocol layer; and

(c) transmitting the header separately from the bitstream received from the step (b).

2.

A method of transmitting a bitstream in a communication network, the method comprising:

(a) coding a source into a bitstream using a predetermined type of coding; and

(b) adding the header of each communication protocol layer to a payload while transmitting the bitstream coded in the step (a) to each communication protocol layer, and separately transmitting the payload and the header.

3.

The method of claim 1, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and only the header information in the bitstream is separately transmitted in an acknowledged or unacknowledged mode protocol.

4.

The method of claim 2, wherein a payload in a bitstream, which has passed through each communication protocol layer, is transmitted in an unacknowledged mode protocol, and the added header is separately transmitted in an acknowledged mode protocol.

5.

The method of claim 3 or 4, wherein, when the number of times of re-transmission of a bitstream transmitted to an acknowledged mode protocol is equal to or less than a predetermined number of times, the bitstream, which has been transmitted in an unacknowledged mode protocol, is transmitted in an acknowledged mode protocol.

6.

The method of claim 3 or 4, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and simultaneously the header information in the bitstream is also transmitted in the acknowledged mode protocol.

7.

The method of claim 3, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and simultaneously the header information in the bitstream is also transmitted in the unacknowledged mode protocol.

8.

The method of claim 3, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and, when a transmission failure occurs, the bitstream is re-transmitted in an acknowledged or unacknowledged mode protocol.

9.

The method of any of claims 3 or 4, wherein the acknowledged mode protocol is a transmission control protocol (TCP), and the unacknowledged mode protocol is a user datagram protocol (UDP).

10.

The method of any of claims 3 or 4, wherein the acknowledged mode protocol

re-transmits data of individual Internet protocol (IP) packets or individual radio link layer protocols (RLP).

11.

The method of claim 1 or 2, wherein the headers are a multimedia header, a real time protocol (RTP) header, a user diagram protocol (UDP) or transmission control protocol (TCP) header, an Internet protocol (IP) header, a radio link layer protocol (RLP) header, and a layer 2 (L2) header which are added after a bitstream is passed through each layer.

12.

The method of claim 1 or 2, wherein the payload includes multimedia data.

13.

An apparatus for transmitting a bitstream in a communication network, the apparatus comprising:

an encoder for encoding source data into a bitstream using a predetermined type of coding;

a protocol processing unit for adding the header of each communication protocol layer to a payload while transmitting the bitstream encoded by the encoder to each communication protocol layer; and

a packet processing unit for transmitting the bitstream processed by the protocol processing unit in an unacknowledged mode protocol and transmitting the header information in an unacknowledged or acknowledged mode protocol.

14.

An apparatus for relaying and receiving a bitstream in a communication network, the apparatus comprising:

an extractor for separately extracting 9 payloads and header information, which corresponds to the header of each layer, while transmitting a bitstream received in a separate transmission protocol in the communication network to each layer;

an error determination processing unit for determining whether the header information extracted by the extractor has error, and requesting re-transmission if it is determined that the header information has error;

a bitstream re-organizing unit for re-organizing a bitstream using the header information extracted by the extractor; and

a decoder for decoding a bitstream re-organized by the bitstream re-organizing unit.

FIG. 1

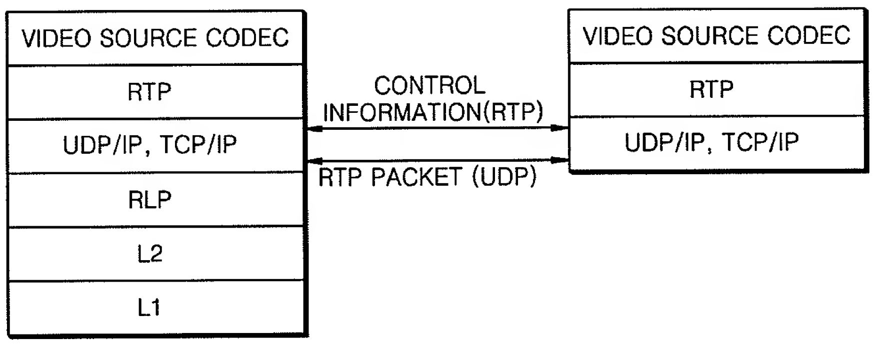

VIDEO SOURCE CODEC
RTP
UDP/IP, TCP/IP
RLP
L2
L1
CONTROL INFORMATION(RTP)
RTP PACKET (UDP)
VIDEO SOURCE CODEC
RTP
UDP/IP, TCP/IP

FIG. 2

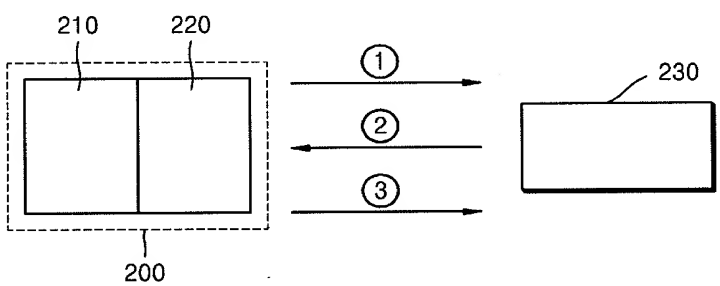

210
220
①
②
③
230
200

FIG. 3

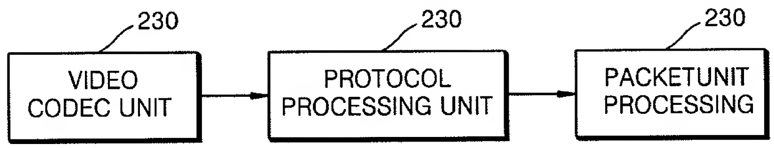

230
VIDEO CODEC UNIT
230
PROTOCOL PROCESSING UNIT
230
PACKETUNIT PROCESSING

FIG. 4

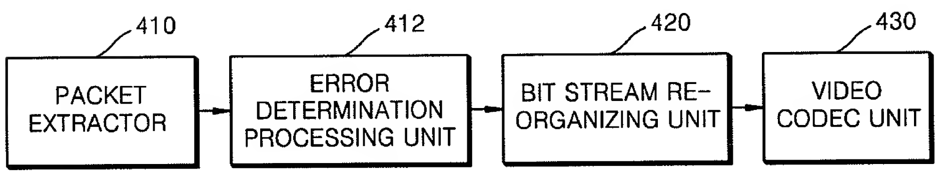
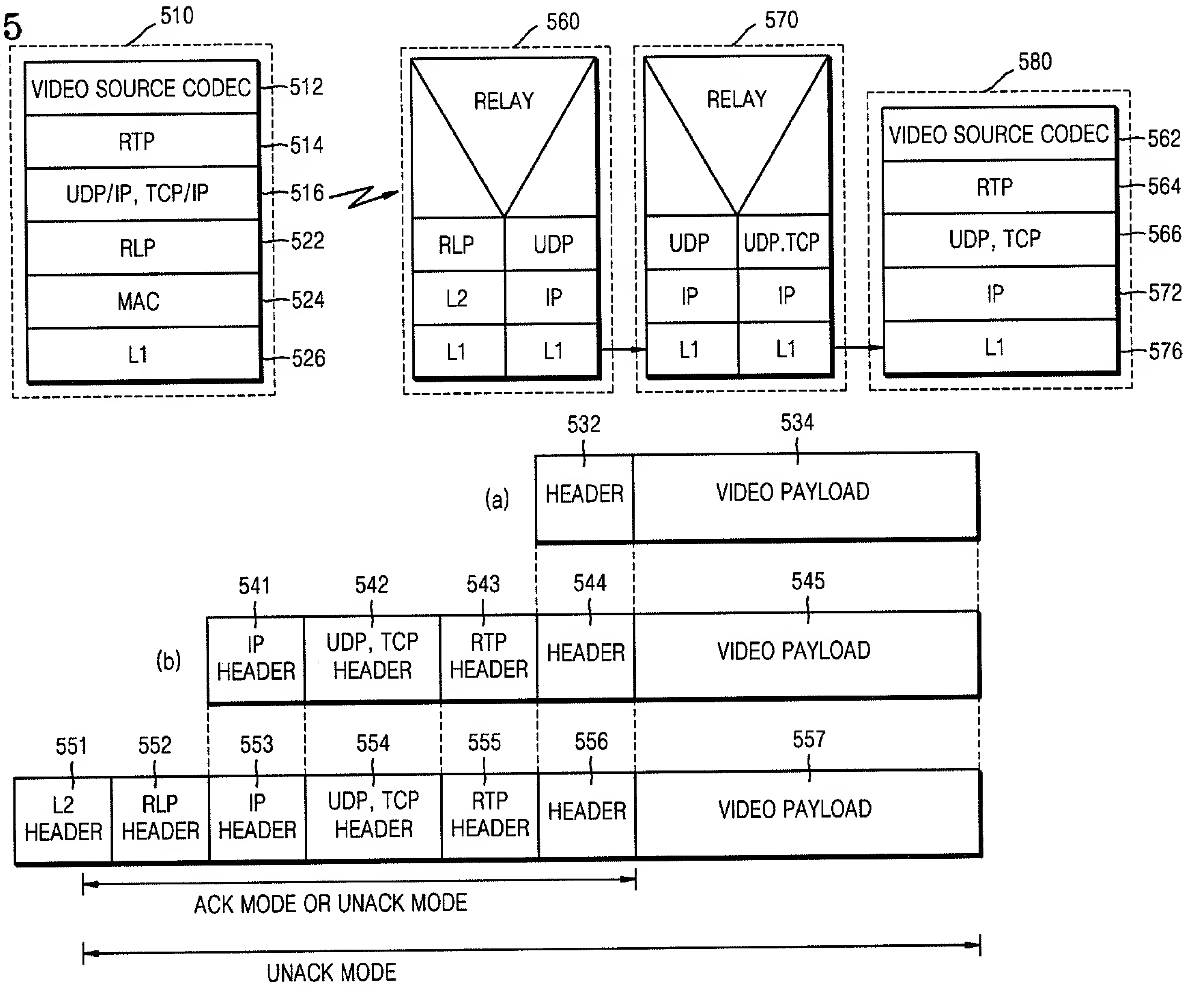

410
PACKET EXTRACTOR
412
ERROR DETERMINATION PROCESSING UNIT
420
BIT STREAM RE-ORGANIZING UNIT
430
VIDEO CODEC UNIT

FIG. 5



510
VIDEO SOURCE CODEC
RTP
UDP/IP, TCP/IP
RLP
MAC
L1
512
514
516
522
524
526
560
RELAY
RLP
UDP
L2
IP
L1
L1
570
RELAY
UDP
UDP.TCP
IP
IP
L1
L1
580
VIDEO SOURCE CODEC
RTP
UDP, TCP
IP
L1
562
564
566
572
576
(a)
532
534
HEADER
VIDEO PAYLOAD
(b)
541
542
543
544
545
IP HEADER
UDP, TCP HEADER
RTP HEADER
HEADER
VIDEO PAYLOAD
(c)
551
552
553
554
555
556
557
L2 HEADER
RLP HEADER
IP HEADER
UDP, TCP HEADER
RTP HEADER
HEADER
VIDEO PAYLOAD
ACK MODE OR UNACK MODE
UNACK MODE